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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/808,694	03/14/2001	Zezhang Hou	AUD1P005	5136

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EXAMINER	
MICHALSKI, JUSTIN I	

ART UNIT	PAPER NUMBER
2644	5

DATE MAILED: 04/09/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 10/03)

Office Action Summary	Application No.	Applicant(s)	
	09/808,694	HOU, ZEZHANG	
	Examiner	Art Unit	
	Justin Michalski	2644	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *14 March 2001*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-20* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-6,9-18 and 20* is/are rejected.

7)☒ Claim(s) *7,8 and 19* is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date *3 and 4*.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.
>
> (e) the invention was described in a patent granted on an application for patent by another filed in the United States before the invention thereof by the applicant for patent, or on an international application by another who has fulfilled the requirements of paragraphs (1), (2), and (4) of section 371(c) of this title before the invention thereof by the applicant for patent.

The changes made to 35 U.S.C. 102(e) by the American Inventors Protection Act of 1999 (AIPA) and the Intellectual Property and High Technology Technical Amendments Act of 2002 do not apply when the reference is a U.S. patent resulting directly or indirectly from an international application filed before November 29, 2000. Therefore, the prior art date of the reference is determined under 35 U.S.C. 102(e) prior to the amendment by the AIPA (pre-AIPA 35 U.S.C. 102(e)).

2. Claims 1, 2, and 10-14 are rejected under 35 U.S.C. 102(b) as being anticipated by Sasaki et al. (Hereinafter "Sasaki") (US Patent 45,471,538).

Regarding Claim 1, Sasaki discloses an adaptive directional sound processing system (Figure 2), comprising: at least first (microphone 11) and second (microphone 21) microphones spaced apart by a distance, said first microphones (11) producing a first electronic sound signal (output of microphone 11) and said second microphone (21) producing a second electronic sound signal (output of microphone 21); means for

processing the second electronic sound signal to adaptively produce a compensation scaling amount (adaptive filter 24) that compensates for sensitivity differences between said first and second microphones (adaptive filter is a function of signals from microphones 11 and 21); a scaling circuit (adaptive filter 24) operatively connected to said means for scaling and said second microphone, said scaling circuit operates to scale the second electronic sound signal in accordance with the compensation scaling amount (function of signals from microphones 11 and 21); and a subtraction circuit (subtractor 15) operatively connected to said scaling circuit (24) and said first microphone (11), said subtraction circuit producing an output difference signal by subtracting the scaled second electronic sound signal from the first electronic sound signal.

Regarding Claim 2, Sasaki further discloses a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount (Sasaki discloses a analog to digital converter 23 which inherently will contain some delay).

Regarding Claim 10, Sasaki discloses a hearing aid device (Figure 2) having an adaptive directional sound processing, said hearing aid device comprising: at least first and second microphones spaced apart by a distance (microphones 11 and 21), said first microphone (11) producing a first electronic sound signal and said second microphone (21) producing a second electronic sound signal; sensitivity difference detection circuitry operatively connected to said first and second microphones (adaptive filter 24), said sensitivity difference detection circuitry adaptively produces a

compensation scaling amount corresponding to sensitivity differences between said first and second microphones (adaptive filter is a function of signals from microphones 11 and 21); a scaling circuit (adaptive filter 24) operatively connected to said sensitivity difference detection circuitry and said second microphone, said scaling circuit operates to scale the second electronic sound signal in accordance with the compensation scaling amount (adaptive filter is a function of signals from microphones 11 and 21); and a subtraction circuit (subtractor 15) operatively connected to said scaling circuit and said first microphone, said subtraction circuit producing and output difference signal by subtracting the scaled second electronic sound signal from the first electronic sound signal.

Regarding Claim 11, Sasaki further discloses a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount (Sasaki discloses a analog to digital converter 23 which inherently will contain some delay).

Regarding Claim 12, Sasaki discloses a method for adaptively measuring and compensating for acoustical differences between sound signals picked up by microphones (Figure 2), said method comprising: (a) receiving first and second electronic sound signals from first and second microphones, respectively (microphones 11 and 21); (b) determining a compensation scaling amount that compensates for acoustic differences with respect to the first and second microphones (control signal from output of subtractor 15 to adaptive filter 24); (c) scaling the second electronic sound signal in accordance with the compensation scaling amount (adaptive filter 24);

and (d) producing a differential electronic sound signal by subtracting the scaled second electronic sound signal from the first electronic sound signal (subtractor 15).

Regarding Claim 13, Sasaki further discloses the acoustic differences pertain to at least differences in microphone sensitivity (subtractor produces difference between signals of microphones 11 and 21).

Regarding Claim 14, Sasaki further discloses determining (b) comprises: (b1) measuring a sensitivity difference between the first and second microphones while in use (it is inherent that subtractor 15 be operational while in use); and (b2) producing the compensation scaling amount based on the sensitivity difference (output from subtractor 15 (i.e. difference) to adaptive filter (i.e. compensation).

3. Claims 1-4, 12-16, and 18 are rejected under 35 U.S.C. 102(e) as being anticipated by Ikeda (US Patent 6,285,768).

Regarding Claim 1, Ikeda discloses an adaptive directional sound processing system (Figure 1), comprising: at least first (microphone terminal 1) and second (microphone terminal 2) microphones spaced apart by a distance, said first microphones producing a first electronic sound signal (y(k)) and said second microphone producing a second electronic sound signal (x(k)); means for processing the second electronic sound signal to adaptively produce a compensation scaling amount that compensates for sensitivity differences between said first and second microphones (adaptive filter 12 is a function of microphones 1 and 2 through subtractor 13); a scaling circuit operatively connected to said means for scaling and said second

microphone (adaptive filter 4), said scaling circuit operates to scale the second electronic sound signal in accordance with the compensation scaling amount (adaptive filter 4 is a function produced from scaling adaptive filter 12); and a subtraction circuit (subtractor 5) operatively connected to said scaling circuit and said first microphone, said subtraction circuit producing an output difference signal (output of 5) by subtracting the scaled second electronic sound signal (output of adaptive filter 4) from the first electronic sound signal.

Regarding Claim 2, Ikeda further discloses a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount (delay circuits 2 and 17).

Regarding Claim 3, Ikeda discloses an adaptive directional sound processing system (Figure 1), comprising: at least first and second microphones spaced apart by a predetermined distance (microphone terminals 1 and 2), said first microphone (1) producing a first electronic sound signal (y(k)) and said second microphone (2) producing a second electronic sound signal (x(k)); a first minimum estimate circuit operatively coupled to said first microphone (power average circuit 14), said first minimum estimate circuit produces a first minimum estimate for the first electronic sound signal from said first microphone (power average circuits 14 and 15 approximates (i.e. estimates the signal power which will inherently include minimum) (Column 4, lines 62-64); a second minimum estimate circuit operatively coupled to said second microphone (power average circuit 15), said second minimum estimate circuit produces a second minimum estimate for the second electronic sound signal from said

second microphone; a divide circuit operatively connected to said first and second minimum estimate circuits (division circuit 16), said divide circuit operates to produce a scaling signal from the first and second minimum estimates (outputs from 15 and 14); a multiply circuit (adaptive filter 4 inherently contains multipliers) operatively connected to said divide circuit (16) and said second microphone (2), said multiply circuit operates to multiply the second electronic sound signal (x(k)) by the scaling signal (output of 19) to produce a scaled second electronic sound signal (r(k)); and a subtraction circuit (subtractor 5) operatively connected to said multiply circuit (4) and said first microphone (1), said subtraction circuit producing an output difference signal by subtracting the scaled second electronic sound signal from the first electronic sound signal.

Regarding Claim 4, Ikeda further discloses a delay circuit that delays the second electronic sound signal or the scaled second electronic sound signal by a delay amount (delay circuits 2 and 17).

Regarding Claim 12, Ikeda discloses a method for adaptively measuring and compensating for acoustical differences between sound signals picked up by microphones (Figure 1), said method comprising: (a) receiving first and second electronic sound signals from first and second microphones, respectively (microphone terminals 1 and 2); (b) determining a compensation scaling amount that compensates for acoustic differences with respect to the first and second microphones (output from subtractor 13 to adaptive filter 12 and input from division circuit 16 to adaptive filter 12); (c) scaling the second electronic sound signal in accordance with the compensation scaling amount (adaptive filter 12); and (d) producing a differential electronic sound

signal by subtracting the scaled second electronic sound signal from the first electronic sound signal (subtractor 13).

Regarding Claim 13, Ikeda further discloses the acoustic differences pertain to at least differences in microphone sensitivity (subtractor 13 produces difference between signals of microphones 1 and 2).

Regarding Claim 14, Ikeda further discloses determining (b) comprises: (b1) measuring a sensitivity difference between the first and second microphones while in use (it is inherent that subtractor 15 be operational while in use); and (b2) producing the compensation scaling amount (output from subtractor 13 to adaptive filter 12) based on the sensitivity difference (output of 13).

Regarding Claim 15, Ikeda further discloses said measuring (b1) of the sensitivity difference is preformed using minimum estimates of the first and second sound signals (power average circuit 14 and 15 approximates, i.e. estimates, the signal power which inherently include minimum) (Column 4, lines 62-64).

Regarding Claim 16, Ikeda further discloses said measuring (b1) f the sensitivity difference is performed using maximum estimates of the first and second sound signals (power average circuit 14 and 15 approximates, i.e. estimates, the signal power which inherently will include maximum) (Column 4, lines 62-64).

Regarding Claim 18, Ikeda further discloses said determining (b) comprises: (b1) determining a first minimum estimate of the first electronic sound signal (power average circuit 14 approximates, i.e. estimates, the signal power which inherently include minimum) (Column 4, lines 62-64); (b2) determining a second minimum estimate of the

second electronic sound signal (power average circuit 15 approximates, i.e. estimates, the signal power which inherently include minimum) (Column 4, lines 62-64); (b3) dividing (division circuit 16) the first minimum estimate by the second minimum estimate to produce the compensation scaling amount (output from 16 to 12).

Claim Rejections - 35 USC § 103

4. Claim 9 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ikeda as applied to claim 3 above in view of Greenberg et al. ("Evaluation of an adaptive beamforming method for hearing aids", J. Acoust. Soc. Am. 91(3), March 1992, pp. 1662-76).

Ikeda discloses a sound processing system as stated apropos of claim 3 above but does not disclose said adaptive directional sound processing system resides within a hearing aid device. Greenberg et al. teaches that multiple microphone adaptive systems in hearing aids are designed to maximize directionality and provide additional benefits in time-varying acoustic conditions (Page 1662, Introduction). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made that adaptive sound processing systems can be used in hearing aid devices in order to provide benefits in time-varying acoustic conditions as taught by Greenberg et al.

5. Claim 20 is rejected under 35 U.S.C. 103(a) as being unpatentable over Sasaki as applied to claim 12 above in view of Greenberg et al. ("Evaluation of an adaptive

beamforming method for hearing aids", J. Acoust. Soc. Am. 91(3), March 1992, pp. 1662-76).

Sasaki discloses a sound processing system method as stated apropos of claim 12 above but does not disclose being performed within a hearing aid device. Greenberg et al. teaches that multiple microphone adaptive systems in hearing aids are designed to maximize directionality and provide additional benefits in time-varying acoustic conditions (Page 1662, Introduction). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made that adaptive sound processing method can be used in hearing aid devices in order to provide benefits in time-varying acoustic conditions as taught by Greenberg et al.

6. Claim 17 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ikeda as applied to claim 14 above in view of Thompson (US Patent 6,654,468).

Ikeda discloses a sound processing method as stated apropos of claim 14 above but does not disclose using Root-Mean-Square averages of the first and second sound signals. Ikeda discloses measuring the difference between the first and second microphones with subtractor 13 in a feedback configuration through average circuits 14 and 15. Thompson discloses a method of controlling a response of a microphone including RMS detectors 26 and 28. Thompson discloses that after detectors 26 and 28 the signals represent the average of the signal. One of ordinary skill in the art at the time the invention was made would have known that an RMS can be used to obtain a type of average power of a signal and could be incorporated in Ikeda circuits 14 and 15

in order to calculate the average of the signals to produce a sensitivity difference between the first and second microphones.

7. Claim 5 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ikeda as applied to claim 3 above in view of Vernon et al. (US Patent 6,268,725).

Ikeda discloses a system as stated apropos of claim 5 above but does not disclose the divide circuit operated in a linear domain. It would have been known at the time the invention was made that a divide circuit can operate in a linear domain as illustrated by Vernon et al. Column 10, lines 40-42.

8. Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Ikeda as applied to claim 3 above in view of Coates (US Patent 4,245,313).

Ikeda discloses a system as stated apropos of claim 5 above but does not disclose the divide circuit operated in a logarithm domain. It would have been known at the time the invention was made that a divide circuit can operate in a logarithm domain as illustrated by Coates Column 15, lines 56-58.

Allowable Subject Matter

9. Claim 7, 8, and 19 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Justin Michalski whose telephone number is (703)305-5598. The examiner can normally be reached on 8 Hours, 5 day/week.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Bill Isen can be reached on (703)305-4386. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

JIM

XU MEI
PRIMARY EXAMINER